Amendments to the Claims:

This listing of claims will replace all prior versions, and listings, of claims in the application:

Listing of Claims:

1. (Currently Amended) A method for coding audio data comprising a sequence of digital audio input samples, including the steps of:

i) multiplying the sequence of digital audio input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients.

2. (Original) A method for coding audio data as claimed in claim 1, wherein the audio coded frequency domain coefficients comprise modified discrete cosine transform coefficients.

3. (Currently Amended) A method for coding audio data as claimed in claim 1, wherein the first trigonometric function factor for each digital audio input sample is a function

of a sequence position of the digital audio input sample ~~sequence position~~ and the number of samples in the sequence.

4. (Previously Presented) A method for coding audio data as claimed in claim 1, wherein the respective second trigonometric function factors for each transform coefficient in the sequence are respective functions of the transform coefficient sequence position and the number of coefficients in the sequence.

5. (Previously Presented) A method for coding audio data as claimed in claim 1, wherein the respective third trigonometric function factors are respective functions of the transform coefficient sequence position.

6. (Currently Amended) A method for coding audio data as claimed in claim 1, wherein step i) comprises multiplying the sequence of digital audio input ~~sequence~~ samples $x[n]$ by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where:

$x[n]$ are the input sequence audio samples;

N is the number of input sequence audio samples; and

$n = 0,....,N-1$.

7. (Previously Presented) A method for coding audio data as claimed in claim 1, wherein step ii) comprises computing the fast Fourier transform of the intermediate sample sequence so as to generate said transform coefficient sequence $G_k = g_{k,r} + jg_{k,i}$, where:

G_k is the transform coefficient sequence;

$g_{k,r}$ are the real transform coefficient components;

$g_{k,i}$ are the imaginary transform coefficient components; and

$k = 0,....,(N/2-1)$.

8. (Previously Presented) A method for coding audio data as claimed in claim 1, wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r} \cos(\pi(k+1/2)/N) - g_{k,i} \sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r} \cos(\pi(k+1/2)/N) + g_{k,i} \sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively.

9. (Previously Presented) A method for coding audio data as claimed in claim 1, wherein steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1 \cos(\pi(2k+1)/4) - T_2 \sin(\pi(2k+1)/4)$$

where Xk are the audio coded frequency domain coefficients; and

$\cos(\pi(2k+1)/4)$ and $\sin(\pi(2k+1)/4)$ are the third trigonometric function factors.

10. (Currently Amended) A method for coding audio data, including the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

<u>processing the complex sample sequence by multiplying the input sequence samples by a first trigonometric function;</u>

determining a Fourier transform coefficient sequence;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

11. (Original) A method for coding audio data as claimed in claim 10, wherein the step of generating first and second transform coefficient sequences comprises, for each corresponding coefficient in the first and second transform coefficient sequences, selecting first and second transform coefficients from said Fourier transform coefficient sequence, determining a complex conjugate of said second transform coefficient, combining said first transform coefficient and said complex conjugate for said first transform coefficient sequence and differencing said first transform coefficient and said complex conjugate for said second transform coefficient sequence.

12. (Previously Presented) A method for coding audio data as claimed in claim 10, wherein the complex sample sequence is processed by multiplying the input sequence samples $z[n]$ by a first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate an intermediate sample sequence, where:

$z[n] = x[n] + jy[n]$ is the complex sample sequence;

$x[n]$ is the first sequence of digital audio samples;

$y[n]$ is the second sequence of digital audio samples;

N is the number of input sequence audio samples in each sequence;

$n = 0,\ldots,N-1$; and

j is the complex constant.

13. (Previously Presented) A method for coding audio data as claimed in claim 11, wherein said first and second transform coefficient sequences are generated according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j$$

where G_k is said first transform coefficient sequence;

G'_k is said second transform coefficient sequence;

N is the number of input sequence audio samples;

$k = 0,\ldots,(N/2-1)$;

Z_k is said first transform coefficient;

Z^*_{N-k-1} is the complex conjugate of said second transform coefficient; and

j is the complex constant.

14. (Previously Presented) A method for coding audio data as claimed in claim 10 further comprising examining said first and second sequences of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

15. (Previously Presented) A method for coding audio data comprising sequences of digital audio samples from a plurality of audio channels as defined in claim 10, further comprising determining a transform length for each of the channels, pairing the channels according to their determined transform length, and coding the audio samples of first and second channels in each pair according to the determined transform length.

16. (Previously Presented) A method for coding audio data as claimed in claim 10, including applying a windowing function in combination with multiplying the complex sample sequence by a first trigonometric function factor.

17. (Previously Presented) A method for coding audio data including the steps of:

obtaining at least one input sequence of digital audio samples;

pre-processing the input sequence samples including applying a pre-multiplication factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence utilizing a fast Fourier transform; and

post-processing the sequence of transform coefficients including applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

18. (Original) A method as claimed in claim 17, wherein the pre-multiplication factor, and first and second post-multiplication factors are trigonometric function factors.

19. (Previously Presented) A method as claimed in claim 17, wherein the pre-multiplication factor applied to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

20. (Previously Presented) A method as claimed in claim 17, wherein the first post-multiplication factors for each transform coefficient in the sequence are trigonometric functions of the transform coefficient sequence position and the number of coefficients in the sequence.

21. (Previously Presented) A method as claimed in claim 17, wherein the second post-multiplication factor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

22. (Previously Presented) A method as claimed in claim 17, wherein the pre-processing operations are performed on each sample in the input sequence individually.

23. (Previously Presented) A method as claimed in claim 17, wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

24. (Previously Presented) A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre-processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence utilizing a fast Fourier transform; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

25. (Original) A method as claimed in claim 24, wherein the pre-multiplication factor for each sample in the complex input sample sequence comprises a complex trigonometric function of the complex input sample sequence position and the number of samples in the sequence.

26. (Previously Presented) A method as claimed in claim 24, wherein the post-processing for each of the first and second channels includes applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

27. (Currently Amended) A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples $x[n]$, $y[n]$ corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence $z[n]$, where $z[n] = x[n] + jy[n]$;

pre-processing the complex input sequence samples including applying a pre-multiplication factor $cos(\pi n/N) + jsin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,....,(N-1)$;

transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k utilizing a fast Fourier transform, wherein $k = 0,....,(N12- 1)$; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0...N/2-1$$
$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0...N/2-1$$

$$X_k = \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N)) - \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N))$$

$$Y_k = \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N)) - \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N))$$

where G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k) = \pi(2k+1)/4$.

28. (Previously Presented) An apparatus for coding an input sequence of digital audio samples comprising:

a pre-transform processor to process the input sequence samples by applying a pre-multiplication factor to obtain modified input sequence samples;

a transform processor to apply a fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

a post-transform processor to process the sequence of transform coefficients by applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of audio coded frequency domain coefficients representing the input sequence of digital audio samples.

29. (Previously Presented) The apparatus of claim 28 wherein the sequence of audio coded frequency domain coefficients are modified discrete cosine transform coefficients

30. (Previously Presented) The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor, and first and second post-multiplication factors applied by the post-transform processor are trigonometric function factors.

31. (Previously Presented) The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

32. (Previously Presented) The apparatus of claim 28 wherein the first post-multiplication factors applied by the post-transform processor for each transform coefficient in the sequence are trigonometric functions of the transform coefficient sequence position and the number of coefficients in the sequence.

33. (Previously Presented) The apparatus of claim 28 wherein the second post-multiplication factor applied by the post-transform processor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

34. (Previously Presented) The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor are applied to each sample in the input sequence individually.

35. (Previously Presented) The apparatus of claim 28 wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

36. (Previously Presented) The apparatus of claim 28 wherein the pre-transform processor is configured to analyze the input sequence of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

37. (Previously Presented) The apparatus of claim 28 wherein the input sequence of digital audio samples comprises first and second sequences of digital audio samples from first and second audio channels;

the pre-transform processor processing the first and second sequences of digital audio samples by combining the first and second sequences of digital audio samples into a single complex sample sequence;

the transform processor applying the fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

the post-transform processor processing the transform coefficient sequence to thereby generate first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from the fast Fourier transform coefficient sequence, the post-transform processor further processing each of the first and second transform coefficient sequences to thereby generate audio coded frequency domain coefficients so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

38. (Previously Presented) The apparatus of claim 37 wherein the pre-transform processor determines a transform length for each of the channels and pairs the

channels according to their determined transform length, the coding the audio samples of first and second channels in each pair being performed based on the determined transform length.

39. (Previously Presented) A apparatus of claim 28, further comprising applying a windowing function in combination with the pre-multiplication factor.